S/N 09/427,815 **PATENT**

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant:	David Rossum	Examiner: Andrew Graham
Serial No.:	09/427,815	Group Art Unit: 2644
Filed:	October 27, 1999	Docket: 2045.009US1
Title:	SAMPLE RATE CONVERTER HAVING DISTRIBUTED FILTERING	

DECLARATION UNDER 37 CFR § 1.132 OF DAVID ROSSUM

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Commissioner for Patents
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I, David Rossum, do hereby declare:

1. I am employed by Creative Technology Ltd. ("Creative") as Chief Scientist. I was the Founder and Chief Technologist of E-Mu Systems, Inc., which was acquired by Creative Technology Ltd. in 1993.

2. I have extensive and pioneering experience in the field of digital audio technology, including being an inventor on 29 issued United States Patents (see Appendix A); an author of numerous publications (see Appendix B); and being involved in development of various pioneering digital audio tools and techniques (see Appendix C).

3. My invention, titled "Sample Rate Converter having Distributed Filtering" (Serial No. 09/427,815), is based on the novel and non-obvious combination of several known techniques, and provides an unexpected beneficial result. In particular, I combine a multipoint variable rate sample rate interpolator with a halfband filter, using the technique of multistage filtering, to form a novel distributed filtering architecture supporting a variable sample rate conversion ratio. This architecture is unlike all prior art, which has always deployed these techniques independently or exclusively. In fact the prior art teaches away from combining these techniques.

4. As stated in the specification of Serial No. 09/427,815 at page 13 lines 16-17, the need met by my invention is a sample rate converter providing high quality and reduced computation

complexity. The computational complexity of a sample rate converter can be quantified by the number of arithmetic operations (additions and multiplications) required to produce an output point. The quality of conversion can be judged by the factors of passband ripple, transition band width, and stopband attenuation of the filter implicit in the sample rate converter. While these three quality parameters can be traded off against each other, it is practical to determine the comparative quality of two converters by a comparison of all three factors. In addition, the sample rate converter must include the flexibility of allowing intended variation of the sample rate conversion ratio to any rational value for each output sample.

5. As an example, my first example embodiment, as shown in Figure 6 of the specification for Serial No. 09/427,815 and described on pages 24-29 of the specification, combines a halfband upsampling filter 602 followed by an multipoint FIR based variable rate interpolator 604. The halfband upsampling filter 602 can be either an IIR or FIR filter.

6. As reported in the specification of Serial No. 09/427,815 at page 12 lines 6-10, prior art Nth order multipoint FIR based interpolators supporting variable sample rate conversion ratios require 2N multiplications and 3N additions per output sample. In particular, as stated in the specification at page 26 lines 9-12, an interpolator of 28^{th} order consumes 56 multiplications and 84 additions per output sample to achieve a the specifications of 0.01dB passband ripple and 96db stopband attenuation with a transition band width appropriate to 48kHz digital audio allowing guardband aliases. Until the advent of my invention, this was the best known technique for producing a high quality variable sample rate conversion ratio sample rate converter having a practical computational complexity.

7. Using the techniques claimed in my application, the same amount of computation produces a startling improvement in performance. Consider first a 7^{th} order multipoint FIR based variable rate interpolator. This requires 14 multiplications and 21 additions per output sample. Yet, as shown in Table 1 on page 22 of the specification for Serial No.

09/427,815, this is able to achieve a passband ripple specification of 0.0001dB and a stopband attenuation specification of 144dB when used in accordance with my specification, paired with an FIR halfband upsampling filter.

8. One appropriate choice is an FIR halfband upsampling filter of order 121. As stated in the specification for Serial No. 09/427,815 at page 24 lines 3-5, an order 4n+1 filter of this type requires n multiplications and 2n additions per input sample. Thus this filter would require 30 multiplications and 60 additions per input sample, and achieve (according to Table 2 on page 23 of the specification) 0.0000002dB passband ripple and 144dB of stopband attenuation. An IIR filter requires even less computational power.

9. The total computational requirement for this embodiment of my invention thus depends on both the input and output sample rates. If the input sample rate is approximately equal to the output sample rate (the sample rate conversion ratio is close to 1), this chosen example will require 14+30=44 multiplications and 21+60=81 additions per output sample, which is less than the 28^{th} order prior art example. Yet the achieved specifications, 144dB of attenuation compared to 96dB, and 0.0001dB of passband ripple compared to 0.01dB, are amazingly superior. Using the prior art technique of a multipoint FIR based interpolator supporting variable sample rate conversion ratios (based on the techniques of Smith and Gossett and known improvements thereto) would require an order of 50, or approximately twice the computation complexity, to achieve comparable specifications. Existing references suggest that there is no substantial improvement in computational load using a multistage approach when the sample rate conversion ratio is near 1. In fact, some references suggest that there may be a loss in computational efficiency in this case. Note that use of a halfband or multistage filter would not be comparable, since these techniques both lack the ability to support variation of the sample rate conversion ratio.

10. Furthermore, in this embodiment, the sample rate is generally being converted to a higher value. This means that the input sample rate is less than the output sample rate, and thus

the efficiency of my invention is even greater. For example, if the input sample rate is ¼ the output sample rate, only 14+7.5 = 21.5 multiplications and 21+15=36 additions are required per output sample. In this case, the required computational complexity for my invention is approximately one fifth of that required for the prior art. While references suggest that there may be improvement in the case of sample rate conversion ratios much less than 1, this is a very substantial improvement, greater than that which would be expected.

11. Analysis shows that the impressive and surprising performance level of my invention compared to what would be expected based on the prior art is found of the other embodiments and other quality levels as well.

10/17/05

Date

David Rossum

APPENDIX A

Issued Patents to inventor David Rossum

Patent No.	Serial No.	Filing Date	Title
6,858,790	10/080527	02/21/2002	Digital sampling instrument employing cache memory
6,622,207	09/654969	09/05/2000	Interpolation looping of prioritized audio samples in cache connected to system bus
6,365,816	09/618963	07/19/2000	Digital sampling instrument employing cache memory
6,324,235	09/084154	05/20/1998	Asynchronous sample rate tracker
6,138,207	08/971238	11/15/1997	Interpolation looping of audio samples in cache connected to system bus with prioritization and modification of bus transfers in accordance with loop ends and minimum block sizes
6,137,043	09/187139	11/06/1998	Digital sampling instrument employing cache memory
6,092,126	08/969312	11/13/1997	Asynchronous sample rate tracker with multiple tracking modes
6,091,269	08/682383	07/17/1996	Method and apparatus for creating different waveforms when synthesizing musical sounds
6,016,522	08/969684	11/13/1997	System for switching between buffers when receiving bursty audio by computing loop jump indicator plus loop start address for read operations in selected buffer
5,943,427	08/425119	04/21/1995	Method and apparatus for three dimensional audio spatialization
5,928,342	08/887100	07/02/1997	Audio effects processor integrated on a single chip with a multiport memory onto which multiple asynchronous digital sound samples can be concurrently loaded
5,925,841	08/903329	07/29/1997	Digital sampling instrument employing cache memory
5,900,570	08/418957	04/07/1995	Method and apparatus for synthesizing musical sounds by frequency modulation using a filter
5,864,876	08/778943	01/06/1997	DMA device with local page table
5,763,800	08/514788	08/14/1995	Method and apparatus for formatting digital audio data
5,698,807	08/611014	03/05/1996	Digital sampling instrument
5,698,803	08/636827	04/23/1996	Digital sampling instrument employing cache memory
5,342,990	07/882178	05/11/1992	Digital sampling instrument employing cache-memory
5,303,309	07/954439	09/30/1992	Digital sampling instrument
5,248,845	07/854554	03/20/1992	Digital sampling instrument
5,170,369	07/576203	08/29/1990	Dynamic digital IIR audio filter and method which provides dynamic digital filtering for audio signals
5,144,676	07/641693	01/16/1991	Digital sampling instrument
5,111,727	07/462392	01/05/1990	Digital sampling instrument for digital audio data
5,072,645	07/462690	01/11/1990	Output stage for a multitimbral electronic musical instrument providing automatic detection of the use of submix outputs

4,987,600	07/390436	08/03/1989	Digital sampling instrument
4,506,579	06/485955	04/18/1983	Electronic musical instrument
4,404,529	06/221299	12/30/1980	Lowpass filter with electronic control of cutoff and resonance
3,986,423	05/531546	12/11/1974	Polyphonic music synthesizer
3,969,682	05/516744	10/21/1974	Circuit for dynamic control of phase shift

APPENDIX B

Publications by author David Rossum

Rossum, Dave, "A Simple Exponential Response Voltage Controlled Oscillator", Electronotes Newsletter, V2, #13, Oct. 10, 1972

Presents a design for a unijunction based oscillator with exponential voltage control.

Rossum, Dave, "Universal Active Filter", Electronotes Newsletter, V2, #15, Nov. 30, 1972

Presents the fundamental architecture for an analog state variable filter.

Rossum, Dave, "New Exponential VCO", Electronotes Newsletter, V2, #16, Dec. 30, 1972

Presents a design for an exponentially voltage controlled oscillator with improved stability and accuracy.

Rossum, Dave, "Synthesizer Keyboards", Electronotes Newsletter, V3, #21, Apr. 30, 1973

Presents the design of a monophonic analog synthesizer keyboard for exponentially controlled circuits.

Rossum, Dave, "On Transient Generators", Electronotes Newsletter, V3, #22, May 20, 1973

Presents an overview of the designs for envelope generators used in music synthesizers.

Rossum, Dave, "More Thoughts on System Design", Electronotes Newsletter, V3, #24, Jun. 21, 1973

Provides guidance on system considerations for the design of modular music synthesizers.

Rossum, Dave, "Polyphonic Synthesis", Electronotes Newsletter, V4, #32, Dec. 15, 1973

Gives an overview of the issues that must be solved in order to create a practical polyphonic music synthesizer.

Rossum, Dave, "TTL in Synthesizer Systems", Electronotes Newsletter, V5, #35, Feb. 15, 1974

Gives an overview of the use of inexpensive computer logic chips in the design of analog music synthesizers.

Rossum, Dave, "On Voltage Controlled Filters", Electronotes Newsletter, V5, #36, Mar 5, 1974

Presents an overview of the designs for voltage controlled filters used in music synthesizers.

Rossum, Dave, "Surplus Parts, Power Supplies and Catastophic Failure", Electronotes Newsletter, V6, #45, Oct. 14, 1974

Describes the potential pitfalls of building hobbyist systems using inexpensive surplus parts, and provides power supply circuitry that is useful in preventing failures from becoming catastrophic.

Rossum, Dave, "The Gilbert Multiplier in Electronic Music", Electronotes Newsletter, V8, #67, July, 1976

Describes the operation of the Gilbert Analog Multiplier and details its uses and limitations in analog electronic music circuits.

Rossum, Dave and Wedge, Scott, "A Microprocessor Based Polyphonic Keyboard for Modular Electronic Music Systems", Proc. 57th Audio Engineering Society Convention, April, 1977

Describes the design details and operation of the first commercial use of a microprocessor for electronic music. The keyboard was not only capable of controlling up to 16 synthesizer channels; the microprocessor and memory allowed the keyboard to memorize performances and play them back, including facilities for editing the stored sequences.

Rossum, Dave, "Two IC's for Electronic Music", Electronotes Newsletter, V9, #78, June, 1977
Describes the design and specifications of two Voltage Controlled Amplifier integrated circuits created specifically for electronic music applications.

Rossum, Dave, "Some Thoughts on Microprocessors in Electronic Music", Computer Music Journal, V1, #2, Summer, 1977
Discusses the benefits and pitfalls regarding the use of microprocessors for the generation of music.

Rossum, Dave and Dow, Ronald, "A Set of Integrated Circuits for Electronic Music", Proc. 60th Audio Engineering Society Convention, April, 1978
Describes the design and specifications of a complete set of integrated circuits created specifically for electronic music applications. Oscillator, filter, amplifier and envelope generation functions are all available.

Rossum, Dave, "A Computer Controlled Polyphonic Synthesizer", Proc. 66th Audio Engineering Society Convention, April, 1980
Describes the Audity, a computer-controlled electronic music system designed for composition, orchestration, and live performance. A live demonstration accompanied the presentation of the paper.

Rossum, Dave, "A Computer Controlled Polyphonic Synthesizer", Journal of the Audio Engineering Society, V29, #12, Dec. 1981
Describes the Audity, a computer-controlled electronic music system designed for composition, orchestration, and live performance.

Rossum, Dave, "The Use of 8 Bit Companding Digital to Analog Converters in Electronic Music", Proc. 72nd Audio Engineering Society Convention, Sep, 1982
Eight bit companding DAC's are generally considered to have insufficient quality to be useful for professional audio. This paper explores techniques through which these inexpensive converters can provide acceptable audio quality.

Rossum, Dave, "Some Aspects of Sample Rate Conversion", Proc. International Conference on Computer Music, August, 1985
Provides a general framework through which both synchronous wavetable synthesis algorithms and asynchronous algorithms can be analyzed, and gives the analysis results for current commercial technologies.

Rossum, Dave, "Digital Musical Instrument Design: The Art of Compromise", The AES 5th International Conference: Music and Digital Technology, Apr. 1987
Describes the design compromises necessary in the design of wavetable synthesizers, with specific examples from commercial products.

Rossum, Dave and Duncan, Andy, "Fundamentals of Pitch-Shifting", Proc. 85th Audio Engineering Society Convention, Oct, 1988
A tutorial on the current state of the art in pitch shifting for wavetable synthesis.

Rossum, Dave, "Digital Dither in Music and Sound Synthesis", The AES 7th International Conference: Audio in Digital Times, April 1989
A tutorial on the current state of the art of digital dither in music synthesis, including a design example from a commercial product.

Rossum, Dave, "An Analysis of Pitch-Shifting Algorithms", Proc. 87th Audio Engineering Society Convention, Sep, 1989

Another tutorial on the current state of the art in pitch shifting for wavetable synthesis. This paper emphasizes the different classes of algebraic interpolators.

Rossum, Dave, "The ARMAdillo Coefficient Encoding Scheme for Digital Audio Filters", IEEE ASSP Workshop on WASPAA, October 1991

Presents a novel coefficient encoding method for digital filters. When linearly interpolated, the resulting dynamic filters are smoothly controllable over a wide range with a pleasing perceptual mapping.

Rossum, Dave, "Making digital filters sound 'analog'", Proc. International Conference on Computer Music, October, 1992

Presents a novel flow diagram for digital filters. The resulting filters respond in a pleasing and natural way to both overload and to rapid variation of the controlling coefficients.

Rossum, Dave, "Constraint Based Audio Interpolators", IEEE ASSP Workshop on WASPAA, October 1993

Presents a novel technique for optimizing the coefficients of higher order interpolators, providing improved performance at a given degree of complexity.

Rossum, Dave, "Creative Spatial Audio", Computer Game Developers Conference, April 24, 1995

Presents the details of Creative's spatial audio API for use in computer games.

Rossum, Dave and Ding, Yinong, "Filter Morphing of Parametric Equalizers and Shelving Filters for Audio Signal Processing", Journal of the Audio Engineering Society, V43, #10, Oct. 1995

The behavior of a second-order digital filter whose coefficients are encoded using the ARMAdillo scheme is studied when it morphs from one frame to another.

Rossum, Dave, "Next Generation Technologies for PC Audio and Music Synthesis", Proc. Computer Game Developers Conference, April, 1997

Presents the details of Creative's newest audio technologies for use in computer games.

Rossum, Dave, "Raising the Quality Bar in PC Audio", WinHEC, April, 1997

Describes the challenges in achieving high fidelity audio on the PC platform, the compelling reasons for meeting these challenges, and some of Creative's proprietary technology that will provide solutions.

Rossum, Dave, "Audio for the 21st Century", COMDEX, November, 1997

Introduces the concepts of Environmental Audio, the capabilities of Creative's implementation, and some details of the underlying hardware support required.

Rossum, Dave, "Trends in PC Audio", Wescon, November, 1997

Describes the future trends in PC audio technology that are expected as the PC user base continues to expand and demand more audio capability.

Rossum, Dave, "Environmental Audio", Proc. Computer Game Developers Conference, May, 1998

Presents the details of Creative's Environmental Audio API for use in computer games, with particular emphasis on how the API allows simplified control for all users regardless of the number of layout of speakers in their particular system.

Rossum, Dave, "An Integrated Approach to Multi-Channel Audio", WinHEC, April, 1999

Examines the nature of multi-channel audio, and advocates analysis from a viewpoint similar to a recording studio. Similarities to matrix algebra are also suggested. The paper compares and contrasts different multi-channel audio formats, then discusses some of the issues involved in fully supporting multi-channel audio input and output.

Rossum, Dave, "Delivery of Digital Audio: Compression and Security", WinHEC, April, 1999

Discusses audio CODEC technology, security mechanisms, and the new Secure Digital Music Initiative standards, and considers their implications for hardware designers and vendors.

APPENDIX C

Contributions to the Field of Audio Technology of David Rossum

1973-1977 – Designed the E-mu modular synthesizer system, which proved to be the most stable and reliable analog music synthesizer ever produced.

1974 – Designed the first practical polyphonic keyboard for music synthesizers.

1976-1977 – Lead designer for the E-mu Microprocessor Keyboard, the first commercial use of a microprocessor for electronic music.

1978-1979 – Co-designed the first custom integrated circuit chipset for music synthesizers.

1979 – Designed the Audity, a studio quality computer controlled music synthesizer, and consulted on the design of the Sequential Circuits Prophet, a semi-professional computer controlled music synthesizer.

1980-1981 – Designed the Emulator, the first practical wavetable music synthesizer.

1982-1983 – Designed the Drumulator, the first microcoded commercial digital music drum machine.

1983-1984 – Led the design team for the Emulator II, the wavetable synthesizer that revolutionized the music industry.

1985-1986 – Designed the "E-chip", a high quality single chip wavetable synthesizer.

1987 – Led the design team of the Emulator III, the first 16 channel true 16 bit quality wavetable synthesizer.

1988-1989 – Developed the underlying technology and designed the "G-chip", the first multichannel multipoint interpolating wavetable synthesis engine, and the basis for E-mu's highly successful "Proteus" product family.

1989-1990 – Developed the underlying technology and designed the "H-chip", the first high quality multichannel synthesizer digital filter chip, and the basis for all of E-mu and Creative's all-digital wavetable synthesizer filters to the present.

1990-1992 – Developed the underlying technology and designed the "G-chip II", the first wavetable synthesis engine employing cache memory, and the basis for all of E-mu's wavetable products beginning with the Emulator IV.

1992-1993 – Led the design team for the EMU8000, the first multipoint interpolating wavetable synthesizer for multimedia applications, and the basis of Creative's SoundBlaster AWE product family.

1993-1994 – Developed the underlying technology for Creative's Quadradic Modulation (CQM) synthesis method, and designed a single chip implementation used in all subsequent SoundBlaster products.

1995-1996 – Led the chip design team that produced the integrated circuits supporting the E-mu System "Darwin" Professional Hard Disk Multitrack Recorder.

1996-1998 – Led the chip design team that produced the EMU10K1, the integrated circuit that, until recently, powered all of Creative's hardware based sound cards and modules.

1998-1999 – Conceived and led the development effort for "Environmental Audio," Creative's integrated multichannel 3D audio technology.

2000-2005 – Provided technical leadership to the chip design team that produced the EMU20K1 ("X-fi"), the integrated circuit powering Creative's recently introduced flagship products.

2000-2005 – Provided technical leadership to several other Creative teams working on projects that remain proprietary to date.